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## MPEG-2 VIDEO TRANSMISSION USING THE HIPERLAN/2 WLAN STANDARD

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### ABSTRACT

*At present, Wireless Local Area Networks (WLANs) supporting broadband multimedia communication are being developed and standardized around the world in the 5GHz band. Systems such as IEEE802.11a and HIPERLAN/2 will provide data rates up to 54 Mbps. Highly compressed data such as video are extremely sensitive to the errors that are commonly encountered in wireless channels. This paper addresses the problems of transmitting MPEG-2 video over WLANs, and in particular HIPERLAN/2. Error resilient video transcoding is proposed as a means of handling channel errors.*

### 1. INTRODUCTION

Recently, there has been an increasing interest in multimedia communications such as the transmission of video, images, and data over wireless broadband access networks. Wireless Local Area Networks (WLANs) provide wideband wireless connectivity between PCs, laptops, and other equipment in corporate, public and home environments. HIPERLAN/2 [1] and IEEE 802.11a are two wireless LAN standards that will operate in the 5GHz band and provide data rates up to 54 Mbps.

Close cooperation between ETSI and IEEE has ensured that the physical layers (Figure 1) of the two standards are harmonized to a large extent [2]. The main differences between IEEE 802.11a and HIPERLAN/2 occur in the Medium Access Control (MAC) [2,3]. The IEEE 802.11 standardization group has specified a common MAC mechanism for IEEE 802.11, IEEE 802.11a, and IEEE 802.11b that

is based on Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) [3].

In HIPERLAN/2 the medium access is based on a TDD/TDMA approach using a MAC frame with a period of 2 ms. The control is centralized to an 'Access Point' (AP) which informs the 'Mobile Terminals' (MTs) at which point in time in the MAC frame they are allowed to transmit their data.

The centralized MAC in HIPERLAN/2 will offer Quality of Service (QoS) support, something that will be difficult with the Ethernet type MAC of 802.11a. However a common MAC between ETSI and IEEE is currently under consideration [4].

Highly compressed data are extremely sensitive to bit or burst errors that are commonly encountered in wireless channels. In HIPERLAN/2 a selective repeat ARQ scheme has been chosen for error control. HIPERLAN/2 also supports an unacknowledged (unreliable, low latency transmission) and a repetition mode that will be employed in broadcast and multicast scenarios. Unicast data can be sent using either acknowledged or unacknowledged mode. It is desirable to limit the use of ARQ in video transmissions in order to avoid network congestion and jitter.

This paper investigates the transmission of the MPEG-2 video bitstream over the HIPERLAN/2 standard. The link adaptation mechanism is examined for perfect link adaptation (transmission mode with the highest throughput), and for link adaptation that takes into account the delay and throughput requirements of an application.

In order to handle channel errors, one alternative to ARQ protocols is the use of error resilient video

coding [5]. To achieve this a video transcoding technique based on EREC [6] is proposed, which minimizes the requirement for ARQ and offers good subjective quality and throughput.

The strategy employed in the video transcoder is to partition the MPEG-2 video stream into low and high priority packetized bit streams. The high priority bit stream is adequately protected by means of ARQ protocols while the low priority bit stream does not employ ARQ in order to avoid network congestion and jitter. Error resilience is achieved by employing EREC [6] in the low priority data.

One of the most serious forms of error propagation within an image codec is caused by the loss of block synchronization (see horizontal strips in Figure 7b). The EREC (Error Resilient Entropy Coding) process overcomes this problem by rearranging the variable length blocks of data, so that these can be packed into a set of fixed-length slots.

For cases where ARQ cannot be employed (for example in multicast transmission) or when its use must be limited, a transparent mode (that accepts all the packets) will be examined. The HIPERLAN/2 standard was designed for wireless computing and generic data communications, where it is assumed that any error in the received data is unacceptable at the application level. The HIPERLAN/2 modem will not normally accept packets containing errors. In order to demonstrate the potential benefits of accepting erroneous packets, a comparison of the performance achieved when corrupted packets are either discarded or received will be presented.

This paper is organized as follows: In Section 2, an overview of MPEG-2 is presented. The HIPERLAN/2 standard together with performance results and the channel models that have been specified for evaluation are presented in Section 3. The link adaptation mechanism is described in Section 4. In Section 5, the video transcoding method employed together with performance results is presented. Finally, a comparison of the performance achieved when corrupted packets are

either discarded or received is given in Section 6. Section 7 discusses the results and Section 8 concludes the paper.

## 2. MPEG-2 BASICS

The MPEG-2 video coding standard [7] is being extensively used for the provision of video services worldwide. It is designed to provide distribution quality television (typically using Main Level, Main Profile) at a bit rate between 4 and 9 Mbit/s depending on scene content.

The MPEG standard relies on two basic techniques: intraframe discrete cosine transform (DCT) for the reduction of spatial redundancy, and interframe motion compensation for the reduction of temporal redundancy. The whole video sequence can be divided into groups of pictures, and each group of pictures consists of one or more frames which use three different encoding types: Intra (I) coded frames which are encoded independently, Predictive (P) frames which are coded using prediction from the previous I and P frames, and Bi-directional (B) frames in which motion compensated prediction is achieved by preceding and upcoming I or P frames.

Intraframe coding is performed on the block level. Each block is transformed from the spatial domain to the frequency domain by DCT. The frequency coefficients are then quantized with a quantization matrix. The quantization coefficients are organized in a zigzag order before variable length coding is applied.

Motion compensation is applied at the macroblock level, and the compressed stream consists of motion vectors and predicted errors.

## 3. HIPERLAN/2 WLAN STANDARD

HIPERLAN/2 [1], defined by ETSI BRAN, will support multiple transmission 'modes' (see Table 1), providing data rates up to 54 Mbps where channel

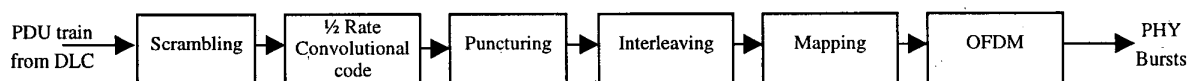


Figure 1: HIPERLAN/2 Transmitter

conditions permit. Thus, it will offer the throughput that is necessary to meet the requirements for multimedia applications like video transmission.

The HIPERLAN/2 radio network is defined in such a way that there are core independent Physical (PHY) and Data Link Control (DLC) layers as well as a set of convergence layers (CL) for interworking with Ethernet, PPP-IP, ATM, UMTS, and IEEE 1394 infrastructure.

### 3.1. HIPERLAN/2 PHY Layer

The physical layer of HIPERLAN/2 is based on the use of Orthogonal Frequency Division Multiplexing (OFDM). OFDM is used to combat frequency selective fading and to randomize the burst errors caused by a wideband-fading channel. The physical layer modes (Table 1) with different coding and modulation schemes are selected by a link adaptation scheme [8,9,11]. Figure 1 shows the reference configuration of the transmitter.

**Table 1: PHY Layer Modes**

| Mode | Modulation | Coding Rate R | Bit rate (Mbit/s) |
|------|------------|---------------|-------------------|
| 1    | BPSK       | 1/2           | 6                 |
| 2    | BPSK       | 3/4           | 9                 |
| 3    | QPSK       | 1/2           | 12                |
| 4    | QPSK       | 3/4           | 18                |
| 5    | 16QAM      | 9/16          | 27                |
| 6    | 16QAM      | 3/4           | 36                |
| 7    | 64QAM      | 3/4           | 54                |

The data is input to a scrambler that prevents long runs of 1s and 0s in the input data being input to the remainder of the modulation process. The scrambled data is input to a convolutional encoder. The encoder consists of a  $\frac{1}{2}$  rate mother code and subsequent puncturing. The puncturing schemes facilitate the use of the following code rates: 1/2, 3/4, 9/16.

The coded data is then interleaved in order to prevent error bursts within the convolutional decoder. The interleaved data is subsequently mapped to data symbols according to either a BPSK, QPSK, 16-QAM or 64-QAM scheme. The OFDM modulation is implemented by means of an inverse

FFT. 48 data symbols and 4 pilots are transmitted in parallel in the form of one OFDM symbol.

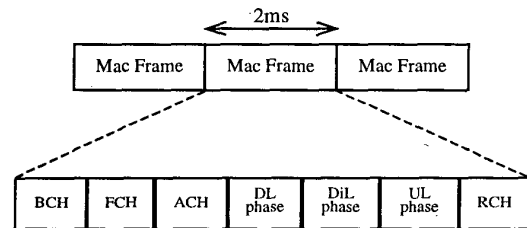
In order to prevent ISI, a guard interval is implemented by means of a cyclic extension. When the guard interval is longer than the excess delay of the radio channel, ISI is eliminated.

The OFDM receiver basically performs the reverse operations of the transmitter. However, the receiver is also required to undertake AGC, time and frequency synchronization and channel estimation. Training sequences are provided in the preamble for the specific purpose of supporting these functions. Two OFDM symbols are provided in the preamble in order to support the channel estimation process.

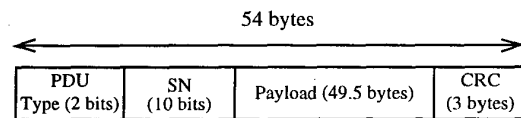
### 3.2. HIPERLAN/2 MAC Layer

In HIPERLAN/2 the Medium Access Control (MAC) is based on a TDD/TDMA approach using a MAC frame with a period of 2 ms. The MAC frame structure (Figure 2) comprises time slots for broadcast control (BCH), frame control (FCH), access feedback control (ACH), and data transmission in downlink (DL), uplink (UL), and directlink (DiL) phases, which are allocated dynamically depending on the need for transmission resources [10].

Downlink, uplink and directlink phases consist of two types of PDUs: long PDUs and short PDUs. Long PDUs (Figure 3) have a size of 54 bytes and contain control or user data.



**Figure 2. The HIPERLAN/2 MAC frame**



**Figure 3: Format of the long PDUs**

### 3.3. Channel Models

HIPERLAN/2 will be deployed in a wide range of environments such as offices, industrial buildings, exhibition halls or even home environments. Different channel models have been produced for the different environments.

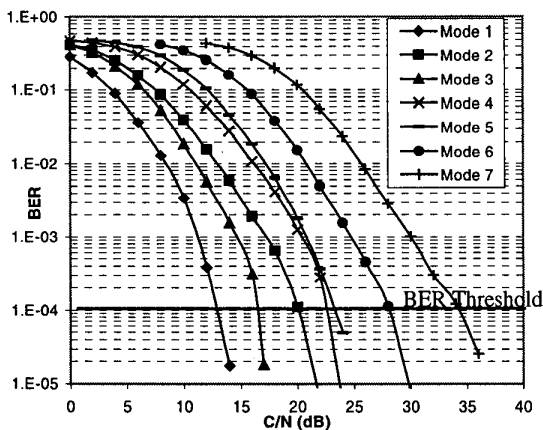
**Table 2: Channel Models**

| Name | RMS delay spread | Characteristic | Environment            |
|------|------------------|----------------|------------------------|
| A    | 50 ns            | Rayleigh       | Office NLOS            |
| B    | 100 ns           | Rayleigh       | Open space/Office NLOS |
| C    | 150 ns           | Rayleigh       | Large Open space NLOS  |
| D    | 140 ns           | Rician         | Large Open space LOS   |
| E    | 250 ns           | Rayleigh       | Large Open space NLOS  |

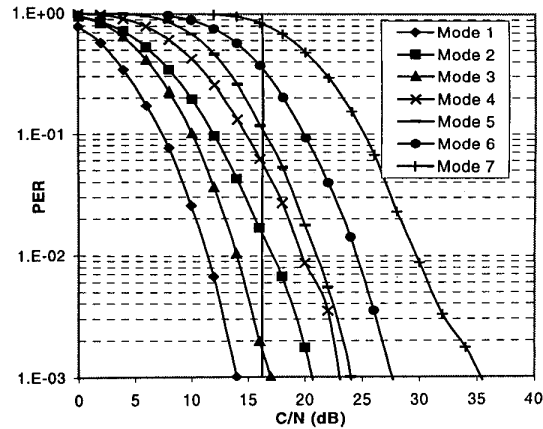
Table 2 summarizes the channel models that were specified [12] for the standard and also used to perform the simulations presented in this paper. The channels are wideband, with Rayleigh or Rician modelled tapped delay lines. Each tap suffers independent Rayleigh or Rician fading with a mean corresponding to an exponentially decaying average power delay profile.

### 3.4. HIPERLAN/2 Performance Results

A fully compliant HIPERLAN/2 modem has been developed [9]. Figures 4 and 5 present the BER and PER performances of HIPERLAN/2 respectively for channel model A.



**Figure 4: BER Performance of HIPERLAN/2**



**Figure 5: PER Performance of HIPERLAN/2**

## 4. LINK ADAPTATION

The physical layer modes (Table 1) with different coding and modulation schemes are selected by a link adaptation scheme. Link adaptation schemes may use a variety of link quality measurements such as PER (Packet Error Rate), received signal strength etc. [8,9,11]. The exact mechanism of this process is not specified in the standard and is left for the vendor to specify. Link adaptation is expected to be performed according to the Quality of Service (QoS) required from each user, i.e. depending on the delay and throughput requirements of the application.

Each packet (PDU) uses CRC-r (Cyclic Redundancy Check) block codes for error detection, where  $r=24$  or 16 for HIPERLAN/2. If a packet is detected to be erroneous by the CRC codes then the terminal will retransmit the packet.

A first order approximation of the link throughput when retransmission is employed is given by:

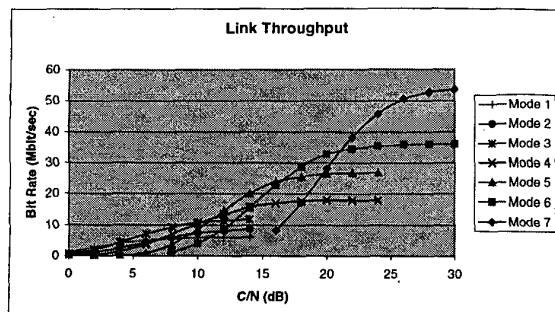
$$\text{Throughput} = R (1 - \text{PER})$$

where  $R$  and  $\text{PER}$  are the bit rate and packet error rate for a specific mode respectively.

In the case of perfect link adaptation, the mode with the highest throughput would be chosen for each instantaneous C/N value [8,9,11].

Figure 6 shows the link throughput in HIPERLAN/2 based on Figure 5 (for PER) and Table 1 (for R) for

the case of perfect link adaptation. The link adaptation mechanism enables the system to adapt the transmission mode to the radio link quality.



**Figure 6: Link Throughput for HIPERLAN/2 (channel model A)**

However in the case of poor radio links the transmission delay increases. Retransmissions must also be considered as additional overheads that reduce the net system capacity.

In [13] the influence of ARQ retransmissions on the transmission delay was analyzed using computer simulations of the DLC (Data Link Control Layer). It was shown that for a C/N value of 16dB transmission delays of 5ms, 6ms, 8ms, 10ms, 15ms, and 25ms were expected for modes 1, 2, 3, 4, 5, and 6 respectively.

This implies that although in an application with no delay constraints at C/N = 16dB the link adaptation scheme would change from mode 5 to mode 6, which has better throughput (see Figure 6), in real time applications this is not the case. By changing mode the transmission delay increases due to the higher PER of this mode for the same C/N value (see Figure 5).

For values of C/N > 30 dB the PER is low and ARQ has little impact on delay or throughput. As the C/N value decreases, the ARQ overhead increases and so the effect on throughput and delay increases.

In [14] it was shown that for perfect link adaptation, packet delays could vary between 5-700 ms (for 1-5 mobile terminals (MTs)) depending on the C/N value and most importantly the number of MTs. The delay was due to queuing and transmission delays.

Congestion, burstiness of errors on the radio link and the delay introduced by the ARQ protocols have to be taken into account in the link adaptation algorithm. Depending on the delay requirements of the applications, link adaptation should choose the mode that optimizes throughput while complying with any required latency constraints.

## 5. VIDEO TRANSCODING

In order to reduce network congestion (as mentioned the delay can reach 700ms for 5MTs) video transcoding is proposed as a means to avoid ARQ for low priority packets.

In a video transcoding scheme the data is partitioned into low and high priority bit streams such that the majority of the headers, motion vectors and low frequency image information is given a high priority while the remaining high frequency information is placed in the low priority partition.

Partitioning of the bit stream is controlled by a MPEG control parameter called the *priority break point* (PBP). This 7-bit integer dictates the destination of each type of Huffman variable length coded (VLC) symbol. If the PBP is 0 then all data is placed in the low priority stream. If the PBP is 127 then all data is placed in the high priority stream. For intermediate values, individual VLC symbols are written to either high or low priority streams.

The high priority bit stream (~30% of the traffic) is adequately protected by means of ARQ protocols while the low priority bit stream does not employ ARQ in order to avoid network congestion and jitter. Error resilience is also achieved by applying EREC [6,15] to the low priority data. One of the most serious forms of error propagation within an image codec is caused by the loss of block synchronization. The EREC process rearranges the variable length blocks of data, so that these can be packed into a set of fixed-length slots.

The use of EREC ensures that variable length blocks of data within a slice start at known points in a slotted data structure. Errors become localized within an 8x8-pixel block. This prevents the propagation of errors that could corrupt the whole slice and cause a visible stripe to appear in the image (see Figure 7b).

For the simulations in this paper, a receiver that accepts corrupted packet was assumed when no retransmission is used (low priority data). The results in Section 6 will demonstrate that this strategy leads to a better performance.

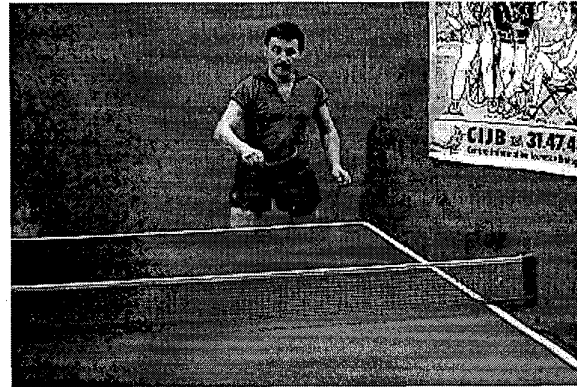
### 5.1. Video Transcoding Performance Results

MPEG-2 data is read from a standard main level main profile (ML@MP) video file containing 4.0Mbit/sec data. The MPEG-2 video test sequence with and without transcoding is then processed through the modem.

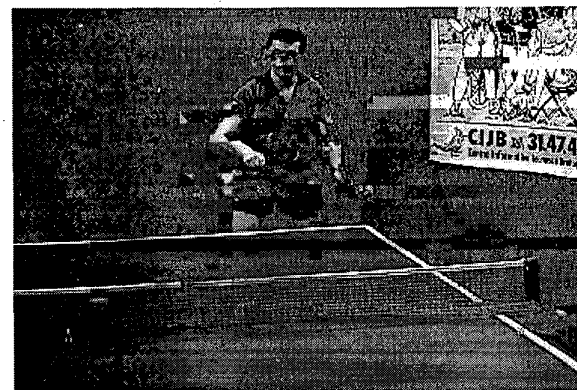
Figure 7 shows example frames at a bit error rate of  $10^{-4}$  (mode 5 in channel A). The quality of the standard MPEG-2 system with ARQ is perfect since retransmissions are employed if a packet is found to contain errors (Figure 7a). The standard MPEG-2 system without ARQ is seen to suffer from a large amount of error propagation, both spatially and temporally (Figure 7b). It can be seen that the system with data partitioning and EREC gives much better subjective quality (Figure 7c). Figure 7 implies that there is a trade-off between quality and delay. The MPEG-2 system with ARQ can provide high quality images (whenever it can be used) but the delay can become very high (see discussion in Section 7); the MPEG-2 system without ARQ can provide low quality, low latency images, while the data partitioning with EREC system provides a trade-off between quality and latency.

Figure 8 shows the measured PSNR values versus BER in channel A for cases b (MPEG-2 without ARQ) and c (data partitioning with EREC). Note that the PSNR in this case measures distortion due to channel errors and not due to compression (i.e. the received video frames are compared with the MPEG-2 frames before transmission).

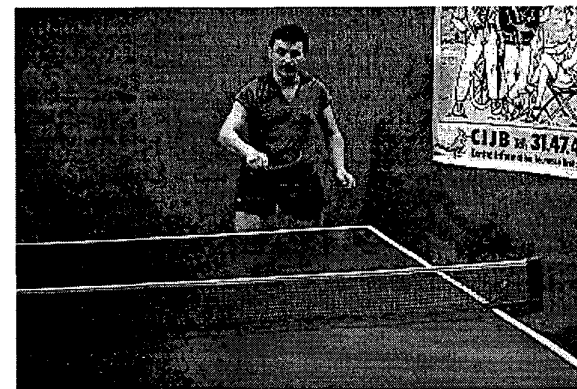
If ARQ is not used for the low priority data, link adaptation can choose the mode with the highest rate for a required BER for each instantaneous C/N value. Hence, by defining a threshold (for example  $\text{BER}=10^{-4}$ , see Figures 7, 8) for the low priority data, the required video quality can be chosen (ex.  $\text{PSNR}>40\text{dB}$ , see Figures 4, 7, 8).



(a)



(b)



(c)

**Figure 7: Example frames ( $\text{BER}=10^{-4}$ ), frame number 115. a) MPEG-2 with ARQ, b) MPEG-2 without ARQ, c) data partitioning with EREC.**

For the high priority data, a PER threshold can be set in order to make sure that the packets will be delivered within an appropriate delay. However, since only the high priority data (less than 30% of the traffic) will use ARQ, delays due to congestion will be minimized.

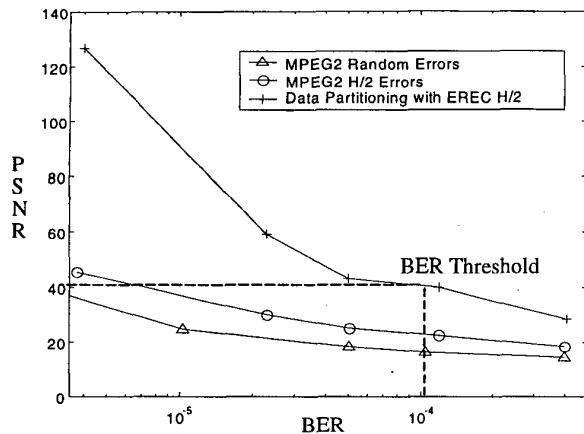


Figure 8: PSNR Performance over BER

Using simple calculations we can estimate the percentage of packets that will require at least one retransmission (delay more than 2ms).

Figure 5 shows the different PERs for the 7 transmission modes, assuming a C/N value of 16dB. If mode 1 is used almost no packet errors occur so no retransmissions need to be performed. If mode 3 is chosen then the  $PER=2 \times 10^{-3}$ , so we expect  $1 - PER=99.8\%$  of the packets to require no retransmission (0.2% require retransmission). For modes 5 and 6 the PER are approximately 10% and 35% respectively. Hence, although the perfect link adaptation scheme of Figure 6 would suggest these modes should be chosen because they provide a higher throughput even with retransmissions, for a video application a lower mode must be employed due to additional delay constraints.

If we define  $\tau$  as the delay due to queuing (depending on the number of MTs), segmentation to HIPERLAN/2 PDUs (which may mean transmission over several MAC frames, depending on the size of the MPEG packets) etc., we can estimate the probability of the delay equalling  $n \cdot (\tau + 2ms)$  as  $1 - PER^n$ . This is, of course, an approximation and a more detailed analysis requires a simulation of the

DLC layer. However, if the delay versus PER is known, link adaptation can choose a transmission mode depending on the joint delay and throughput requirements of the application.

## 6. BENEFITS OF USING CORRUPTED PACKETS

As mentioned earlier, a comparison of the performance achieved when corrupted packets are either discarded or received has been performed in this study. For the cases where retransmissions are not possible (broadcast scenarios), a transparent mode would be very useful.

In our experiments, the data belonging to lost packets is filled with zeros. Figure 9 shows measured PSNR values for a range of packet error rates for the data partitioning case. Similar results were observed for the standard MPEG-2 system.

The HIPERLAN/2 standard was designed for data communications where it is assumed that any error in the received data is unacceptable. The HIPERLAN/2 modem will not normally accept packets containing errors. However, as can be seen from Figure 9, the performance of the system is significantly improved for MPEG-2 transmission when corrupted packets are used. This arises since moderate error rates are acceptable for error resilient video applications since the decoder makes use of data in the corrupted packets.

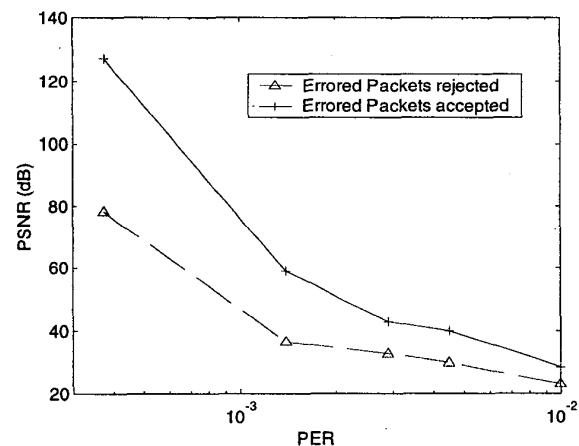


Figure 9: Benefits of using corrupted packets



## 7. DISCUSSION OF RESULTS

Results suggest that we should identify the conditions under which a standard MPEG-2 transmission over HIPERLAN/2 is possible. Two cases will be discussed; when ARQ is employed and when ARQ cannot be used (for example broadcast scenarios).

### 7.1 Transmission with ARQ

For low number of MTs and high values of C/N the PER is low and hence low delay and high capacity is expected. Under these circumstances very high quality MPEG-2 video transmission is expected. The high capacity that the HIPERLAN/2 system offers for high C/N values will ensure robust video transmission when ARQ is employed.

As the number of MTs increases or the C/N decreases, a PER threshold should be defined in order to reduce delay. Depending on the delay requirements of the applications, link adaptation should choose the mode that optimizes throughput while complying with any required latency constraints.

An example of link adaptation with a PER threshold of 1-2% can be seen in Figure 10.

It can be seen that the PER threshold reduces throughput for a specific C/N value since the transition points have moved to the right (compared with Figure 6). It can be observed that a minimum C/N value of 10 dB is necessary to maintain a PER less than 1-2%.

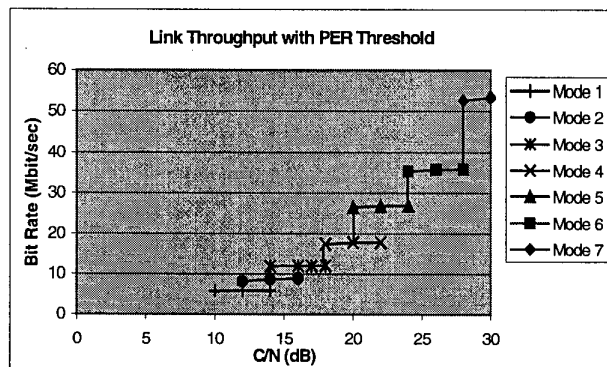


Figure 10: Link Adaptation with PER Threshold

For larger numbers of MTs or for low C/N values, packet delays could reach 700 ms [14]. For this case the system with data partitioning would provide better performance. Since only 30% of the data will use ARQ, congestion can be reduced. This will reduce both the required capacity and delay. The low priority data will be transmitted with no additional delay and a BER threshold can be defined in order to choose the required video quality. The acceptable PER threshold for the high priority data will be higher than the standard MPEG-2 case since only 30% of the data are retransmitted. A transparent mode is also recommended for the packets that are not retransmitted.

### 7.2 Transmission without ARQ

When retransmissions are not possible, the quality of standard MPEG-2 video transmission will be very poor, even if a transparent mode is used. This is due to the fact that errors propagate in the video sequence and reduce image quality significantly. Transmission is possible only if a very low BER threshold is specified and if only the lower modes are used.

The data partitioning system will allow two BER thresholds, one for the high and one for the low priority data. The high priority data will have a similar BER threshold to the standard MPEG-2 system but the low priority data can afford a higher BER rate. Hence capacity will be improved since higher modes can be used for 70% of the traffic.

In all cases the system performance is significantly improved for MPEG-2 transmission when corrupted packets are used.

## 8. CONCLUSIONS

Compressed video has multiple QoS requirements in terms of delay, throughput and error rate. In this paper the transmission of MPEG-2 video over WLANs, and in particular HIPERLAN/2, was investigated.

The PER and BER performances of HIPERLAN/2 were presented over wideband channels. The link adaptation mechanism employed in HIPERLAN/2 enables the system to adapt the transmission mode to

the radio link quality. It was shown that link adaptation should choose the mode that optimizes throughput while complying with the application's latency constraints. In the case of poor radio links the transmission delay increases due to retransmissions.

In order to avoid network congestion, video transcoding was proposed as a means of handling channel errors, and thus reducing the delay caused by ARQ protocols.

In our system only the high priority bit stream is protected by means of ARQ protocols, while the low priority bit stream avoids ARQ in order to reduce network congestion and jitter. Error resilience is also achieved by applying EREC to the low priority data in order to avoid the loss of block synchronization (Figure 7). Figures 7,8 showed that the system with data partitioning and EREC gave much better quality than a standard MPEG-2 system.

It was shown that if ARQ is not used for the low priority data, link adaptation can choose the mode with the highest rate for a required BER for each instantaneous C/N value. For the high priority data, a PER threshold can be set in order to make sure that the packets will be delivered within an appropriate delay.

In order to demonstrate the potential benefits of a transparent mode in HIPERLAN/2, the effects of using corrupted packets were simulated. Results showed (Figure 9) that the performance is significantly improved. This will be particularly useful in cases where retransmissions are not possible.

## ACKNOWLEDGEMENTS

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